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EXAMINER	
LIU, BEN H	

ART UNIT	PAPER NUMBER
2416	

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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	Application No.	Applicant(s)	
	10/811,227	STERNERSON, HANNES	
	Examiner	Art Unit	
	BEN H. LIU	2416	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *03 March 2009*.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1,2,4-7,9-15 and 17-23* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1,2,4-7,9-15 and 17-23* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____.

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Response to Amendment

1. This is in response to an amendment/response filed on March 3rd, 2009.

2. Claims 1, 7, 11, 15 and 20 have been amended.

3. No claims have been cancelled.

4. No claims have been added.

5. Claims 1-2, 4-7, 9-15, and 17-23 are currently pending.

Claim Rejections - 35 USC § 103

6. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

7. The factual inquiries set forth in *Graham* v. *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

1. Determining the scope and contents of the prior art.
2. Ascertaining the differences between the prior art and the claims at issue.
3. Resolving the level of ordinary skill in the pertinent art.
4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

8. **Claims 1-5, 7, 9-10, 11-13, 15, 17-19, and 20-23** are rejected under 35 U.S.C. 103(a) as being unpatentable over Benyassine et al. (U.S. Patent 6,721,712) in view of Kramer et al. (U.S. Patent 6,658,027).

For independent claim 1, Benyassine et al. disclose a system for providing frame rate conversion for audio data comprising a first client configured to transmit audio data frames at a first frame rate *(see figure 1, which recite a client 120 that transmits speech data frames at a first transmission rate)*, a second client configured to receive audio data frames at a second frame rate, wherein the first frame rate is different from the second frame rate *(see figure 1, which recite a client 140 that receives speech data frames at a different rate from the original transmission rate)*; and a device configured to facilitate transmission of audio data frames between the first client and the second client *(see figure 1, which recite a conversion module 130 that facilitates the transmission of frames between the transmitter and receiver clients)*, and further configured to repackage the audio data frames into one or more frames for transmission to the second client at the second frame rate *(see column 5 lines 62-67, column 6 lines 1-2, and figure 1, which recite a rate decoding module 132 that repackages encoded speech signal 101A into converted speech signal 101B)*,

and wherein a total amount of audio data received by the second client in the one or more repackaged frames is equal to a total amount of audio data transmitted by the first client in the audio data frames *(see column 5 lines 62-67 and column 6 lines 1-2, which recite the conversion module 130 that receives frames containing speech audio data and transmitting the speech audio data to receiver 140. Since all the speech audio*

data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client).

Benyassine et al. disclose the subject matter of the claimed invention with the exception wherein the device that facilitates transmission of audio data is configured to store the audio data frames received from the first client in an intermediate storage area. Kramer et al. from the same or similar fields of endeavor disclose a method and system wherein a receiver does not receive a transmission at the same rate it was transmitted *(see column 3 lines 53-55).* The system uses a buffer 120 in a VoIP apparatus 100 to temporarily store the frames for processing *(see figure 1 and column 3 lines 55-67).* Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that stores frames in an intermediate storage area as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. in order to store received frames. The motivation for using the intermediate storage area with the method and system for providing frame rate conversion for audio data is to improve the resilience of the system against variable transmission delays.

Benyassine et al. disclose all the subject matter of the claimed invention with the exception wherein the audio data flames transmitted at the first flame rate have a first interval between the frames, wherein the audio data flames transmitted at the second frames rate have a second interval between the frames, and wherein the first interval and

the second interval are constant. However, even if speech-data is suspended during a pause in the conversation, Kramer et al. from the same or similar fields of endeavor disclose a method and system that uses a jitter buffer 120 *(see column 3 lines 53-61)* that stores transmitted frames and outputs the stored frames at a constant output rate *(see column 1 lines 25-29)*. The jitter buffer is able to insert silence frames during gaps in speech *(see column 2 lines 11-13)* thus compensating for any suspension of speech-data transmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that inserts silence frames during gaps in speech as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 with the ability to insert silence frames as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. The combination allows for the conversion module 130 as taught by Benyassine et al. to receive frames at a constant interval and also transmit data frames at a constant interval. The motivation for using the jitter buffer that is able to insert silence frames with the method and system for providing frame rate conversion for audio data is to improve the reliability of the system against buffer underflow.

For independent claim 7, Benyassine et al. disclose a Voice-over-IP device for facilitating communications between a first client and a second client the device comprising control logic configured to receive audio data frames from the first client at a first frame rate *(see figure 1, which recite a conversion module 130 that receives speech data frames at a first transmission rate from client 120)*; and control logic to repackage

the stored audio data frames into one or more frames for transmission to the second client at a second frame rate *(see column 5 lines 62-67, column 6 lines 1-2, and figure 1, which recite a rate decoding module 132 that repackages encoded speech signal 101A into converted speech signal 101B)*; and a control logic configured to transmit the one or more repackaged frames to the second client at the second frame rate wherein the first frame rate is different from the second frame rate *(see figure 1, which recite a conversion module 130 that transmits speech data frames at a rate different from the original transmission rate to client 140)*,

and wherein a total amount of audio data received by the second client in the one or more repackaged frames is equal to a total amount of audio data transmitted by the first client in the audio data frames *(see column 5 lines 62-67 and column 6 lines 1-2, which recite the conversion module 130 that receives frames containing speech audio data and transmitting the speech audio data to receiver 140. Since all the speech audio data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client)*.

Benyassine et al. disclose the subject matter of the claimed invention with the exception of a control logic used to store the audio data frames from the first client in an intermediate storage area. Kramer et al. from the same or similar fields of endeavor disclose a method and system wherein a receiver does not receive a transmission at the same rate it was transmitted *(see column 3 lines 53-55)*. The system uses a buffer 120 in a VoIP apparatus 100 to temporarily store the frames for processing *(see figure 1 and column 3 lines 55-67)*. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that stores

frames in an intermediate storage area as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. in order to store received frames. The motivation for using the intermediate storage area with the method and system for providing frame rate conversion for audio data is to improve the resilience of the system against variable transmission delays.

Benyassine et al. disclose all the subject matter of the claimed invention with the exception wherein the audio data flames transmitted at the first flame rate have a first interval between the frames, wherein the audio data flames transmitted at the second frames rate have a second interval between the frames, and wherein the first interval and the second interval are constant. However, even if speech-data is suspended during a pause in the conversation, Kramer et al. from the same or similar fields of endeavor disclose a method and system that uses a jitter buffer 120 *(see column 3 lines 53-61)* that stores transmitted frames and outputs the stored frames at a constant output rate *(see column 1 lines 25-29)*. The jitter buffer is able to insert silence frames during gaps in speech *(see column 2 lines 11-13)* thus compensating for any suspension of speech-data transmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that inserts silence frames during gaps in speech as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer

120 with the ability to insert silence frames as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. The combination allows for the conversion module 130 as taught by Benyassine et al. to receive frames at a constant interval and also transmit data frames at a constant interval. The motivation for using the jitter buffer that is able to insert silence frames with the method and system for providing frame rate conversion for audio data is to improve the reliability of the system against buffer underflow.

For independent claim 11, Benyassine et al. disclose a system for providing frame rate conversion for audio data comprising a first client configured to transmit audio data frames at a first frame rate *(see figure 1, which recite a client 120 that transmits speech data frames at a first transmission rate)*, a second client configured to receive audio data frames at a second frame rate, wherein the first frame rate is different from the second frame rate*(see figure 1, which recite a client 140 that receives speech data frames at a different rate from the original transmission rate)*, and a device configured to repackage the stored audio data frames into one or more frames for transmission to the second client at the second frame rate *(see column 5 lines 62-67, column 6 lines 1-2, and figure 1, which recite a rate decoding module 132 that repackages encoded speech signal 101A into converted speech signal 101B)*,

and wherein a total amount of audio data received by the second client in the one or more repackaged frames is equal to a total amount of audio data transmitted by the first client in the audio data frames *(see column 5 lines 62-67 and column 6 lines 1-2, which recite the conversion module 130 that receives frames containing speech audio data and transmitting the speech audio data to receiver 140. Since all the speech audio*

data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client).

Benyassine et al. disclose the subject matter of the claimed invention with the exception of an intermediate storage area configured to store audio data frames received from the first client. Kramer et al. from the same or similar fields of endeavor disclose a method and system wherein a receiver does not receive a transmission at the same rate it was transmitted *(see column 3 lines 53-55).* The system uses a buffer 120 in a VoIP apparatus 100 to temporarily store the frames for processing *(see figure 1 and column 3 lines 55-67).* Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that stores frames in an intermediate storage area as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. in order to store received frames. The motivation for using the intermediate storage area with the method and system for providing frame rate conversion for audio data is to improve the resilience of the system against variable transmission delays.

Benyassine et al. disclose all the subject matter of the claimed invention with the exception wherein the audio data flames transmitted at the first flame rate have a first interval between the frames, wherein the audio data flames transmitted at the second frames rate have a second interval between the frames, and wherein the first interval and the second interval are constant. However, even if speech-data is suspended during a

pause in the conversation, Kramer et al. from the same or similar fields of endeavor disclose a method and system that uses a jitter buffer 120 *(see column 3 lines 53-61)* that stores transmitted frames and outputs the stored frames at a constant output rate *(see column 1 lines 25-29)*. The jitter buffer is able to insert silence frames during gaps in speech *(see column 2 lines 11-13)* thus compensating for any suspension of speech-data transmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that inserts silence frames during gaps in speech as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 with the ability to insert silence frames as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. The combination allows for the conversion module 130 as taught by Benyassine et al. to receive frames at a constant interval and also transmit data frames at a constant interval. The motivation for using the jitter buffer that is able to insert silence frames with the method and system for providing frame rate conversion for audio data is to improve the reliability of the system against buffer underflow.

For independent claim 15, Benyassine et al. disclose a method for providing frame rate conversion for audio data, the method comprising receiving audio data frames from a first client at a first frame rate *(see figure 1, which recite a conversion module 130 that receives speech data frames at a first transmission rate from client 120)*; converting the received audio data frames into one or more frames *(see figure 1 and column 5 lines 50-67, which recite a rate converting module 134 that converts encoded speech signal*

101A into converted speech signal 101B); and transmitting the one or more frames to a second client at a second frame rate; wherein the first frame rate is different from the second frame rate *(see figure 1, which recite a conversion module 130 that transmits speech data frames at a rate different from the original transmission rate to client 140)*,

and wherein a total amount of audio data received by the second client in the one or more repackaged frames is equal to a total amount of audio data transmitted by the first client in the audio data frames *(see column 5 lines 62-67 and column 6 lines 1-2, which recite the conversion module 130 that receives frames containing speech audio data and transmitting the speech audio data to receiver 140. Since all the speech audio data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client)*.

Benyassine et al. disclose the subject matter of the claimed invention with the exception of storing the received audio data frames in an intermediate storage area. Kramer et al. from the same or similar fields of endeavor disclose a method and system wherein a receiver does not receive a transmission at the same rate it was transmitted *(see column 3 lines 53-55)*. The system uses a buffer 120 in a VoIP apparatus 100 to temporarily store the frames for processing *(see figure 1 and column 3 lines 55-67)*. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that stores frames in an intermediate storage area as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as

taught by Benyassine et al. in order to store received frames. The motivation for using the intermediate storage area with the method and system for providing frame rate conversion for audio data is to improve the resilience of the system against variable transmission delays.

Benyassine et al. disclose all the subject matter of the claimed invention with the exception wherein the audio data flames transmitted at the first flame rate have a first interval between the frames, wherein the audio data flames transmitted at the second frames rate have a second interval between the frames, and wherein the first interval and the second interval are constant. However, even if speech-data is suspended during a pause in the conversation, Kramer et al. from the same or similar fields of endeavor disclose a method and system that uses a jitter buffer 120 *(see column 3 lines 53-61)* that stores transmitted frames and outputs the stored frames at a constant output rate *(see column 1 lines 25-29)*. The jitter buffer is able to insert silence frames during gaps in speech *(see column 2 lines 11-13)* thus compensating for any suspension of speech-data transmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that inserts silence frames during gaps in speech as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 with the ability to insert silence frames as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. The combination allows for the conversion module 130 as taught by Benyassine et al. to receive frames at a constant interval and also transmit data frames at a constant

interval. The motivation for using the jitter buffer that is able to insert silence frames with the method and system for providing frame rate conversion for audio data is to improve the reliability of the system against buffer underflow.

For independent claim 20, Benyassine et al. disclose a method for providing frame rate conversion for audio data, the method comprising receiving audio data frames from a first client at a first frame rate *(see figure 1, which recite a conversion module 130 that receives speech data frames at a first transmission rate from client 120)*; repackaging the stored audio data frames into one or more frames *(see column 5 lines 62-67, column 6 lines 1-2, and figure 1, which recite a rate decoding module 132 that repackages encoded speech signal 101A into converted speech signal 101B)*; and transmitting the one or more frames to a second client at a second frame rate; wherein the first frame rate is different from the second frame rate *(see figure 1, which recite a conversion module 130 that transmits speech data frames at a rate different from the original transmission rate to client 140)*,

and wherein a total amount of audio data received by the second client in the one or more repackaged frames is equal to a total amount of audio data transmitted by the first client in the audio data frames *(see column 5 lines 62-67 and column 6 lines 1-2, which recite the conversion module 130 that receives frames containing speech audio data and transmitting the speech audio data to receiver 140. Since all the speech audio data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client)*.

Benyassine et al. disclose the subject matter of the claimed invention with the exception of storing the received audio data frames in an intermediate storage area.

Kramer et al. from the same or similar fields of endeavor disclose a method and system wherein a receiver does not receive a transmission at the same rate it was transmitted *(see column 3 lines 53-55)*. The system uses a buffer 120 in a VoIP apparatus 100 to temporarily store the frames for processing *(see figure 1 and column 3 lines 55-67)*. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that stores frames in an intermediate storage area as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. in order to store received frames. The motivation for using the intermediate storage area with the method and system for providing frame rate conversion for audio data is to improve the resilience of the system against variable transmission delays.

Benyassine et al. disclose all the subject matter of the claimed invention with the exception wherein the audio data flames transmitted at the first flame rate have a first interval between the frames, wherein the audio data flames transmitted at the second frames rate have a second interval between the frames, and wherein the first interval and the second interval are constant. However, even if speech-data is suspended during a pause in the conversation, Kramer et al. from the same or similar fields of endeavor disclose a method and system that uses a jitter buffer 120 *(see column 3 lines 53-61)* that stores transmitted frames and outputs the stored frames at a constant output rate *(see column 1 lines 25-29)*. The jitter buffer is able to insert silence frames during gaps in

speech *(see column 2 lines 11-13)* thus compensating for any suspension of speech-data transmission. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to implement the system and method that inserts silence frames during gaps in speech as taught by Kramer et al. with the method and system for providing frame rate conversion for audio data as taught by Benyassine et al. The method and system as taught by Kramer et al. can be combined by coupling the buffer 120 with the ability to insert silence frames as taught by Kramer et al. to the rate decoding module 132 of the conversion module 130 as taught by as taught by Benyassine et al. The combination allows for the conversion module 130 as taught by Benyassine et al. to receive frames at a constant interval and also transmit data frames at a constant interval. The motivation for using the jitter buffer that is able to insert silence frames with the method and system for providing frame rate conversion for audio data is to improve the reliability of the system against buffer underflow.

For claim 2, Benyassine et al. disclose a system for providing frame rate conversion for audio data wherein the device is further configured to receive the audio data frames from the first client at the first frame rate and convert the audio data frames for transmission to the second client at the second frame rate *(see figure 1 and column 5 lines 50-67, which recite a rate converting module 134 that converts encoded speech signal 101A into converted speech signal 101B).*

For claims 4, 9, 12, 17, and 21, Benyassine et al. disclose a system and method for providing frame rate conversion wherein the system is implemented in software, hardware or a combination of both *(see column 3 lines 38-43, which recite implementing*

the functional blocks of the system as hardware components and/or software components configured to perform the specified functions).

For claims 5, 10, 13, 19, and 23, Benyassine et al. disclose a system and method for providing frame rate conversion for audio data wherein the first client and the second client include telephonic equipment *(see column 4 lines 15-19, which recite a client vocoder that is a CDMA cellular telephone)* and computers *(see column 1 lines 59-64 and column 5 lines 16-29, which recite a client vocoder computer system configured to operate in G.729 VoIP systems).*

9. **Claims 6, 14, 18, and 22** are rejected under 35 U.S.C. 103(a) as being unpatentable over Benyassine et al. (U.S. Patent 6,721,712) and Kramer et al. (U.S. Patent 6,658,027) as applied to claims 1, 11, 15, and 20 and further in view of Mizusawa et al. (U.S. Patent Application Publication 2002/0037002).

For claims 6, 14, 18, and 22, Benyassine et al. and Kramer et al. disclose all the subject matter of the claimed invention with the exception that a Voice-over-IP gateway is used for facilitating the communications between a first client configured to transmit audio data at a first frame rate and a second client configured to receive audio data at a second frame rate. Mizusawa et al. from the same or similar fields of endeavor disclose a VoIP gateway device for performing frame conversion of frame formats with different bit rates *(see paragraph 9)*. Thus, it would have been obvious to the person of ordinary skill in the art at the time of the invention to use the VoIP gateway for frame conversion as taught by Mizusawa et al. with the system and method for facilitating the communications between a first client configured to transmit audio data at a first frame

rate and a second client configured to receive audio data at a second frame rate as taught by Benyassine et al. and Kramer et al. Thus, the VoIP gateway device for performing frame conversion of frame formats as taught by Mizusawa et al. can configured to perform the function of the conversion module 130 through software as taught by Benyassine et al. and Kramer et al. The VoIP gateway can then be deployed as the conversion module. The motivation for using the VOIP gateway device for performing frame conversion of frame formats as taught by Mizusawa et al. with the system and method for facilitating the communications between a first client configured to transmit audio data at a first frame rate and a second client configured to receive audio data at a second frame rate as taught by Benyassine et al. and Kramer et al. is to improve compatibility of the system by providing VoIP access as well as additional functions of private branch exchanges.

Response to Arguments

10. It is noted with appreciation that the Applicant has carefully considered the previous Office Action and the cited prior art references. However, the Applicant's arguments filed March 3rd, 2009 regarding the 35 USC 103(a) rejections have been fully considered but are not persuasive.

Claims 1-5, 7, 9-10, 11-13, 15, 17-19, and 20-23 were previously rejected under 35 U.S.C. 103(a) as being unpatentable over Benyassine et al. (U.S. Patent 6,721,712), hereinafter referred to as “Benyassine,” in view of Kramer et al. (U.S. Patent 6,658,027), hereinafter referred to as “Kramer.” The Applicant suggests that, "Benyassine describes two possible configurations: (1) a DTX-enabled transmitter communicating with a non-

DTX receiver, and (2) a non-DTX enabled transmitter communicating with a DTX-enabled receiver" *(see Applicant's remarks, page 8 lines 1-3).*

For the first configuration, the Applicant alleges that, "the rate conversion module 130 converts the continuous stream of data from the non-DTX transmitter into a discontinuous stream of data for the DTX-enabled receiver by dropping data for periods where no speech is being transmitted" *(see Applicant's remarks, page 9 lines 9-12).* However, it is noted that the frames that are dropped contain background information such as energy level and spectrum represented *(see Benyassine, column 6 lines 36-39).* In contrast, frames containing speech audio data received by the conversion module 130 are transmitted to the receiver 140 *(see Benyassine, column 5 lines 62-67 and column 6 lines 1-2).* Therefore, using the broadest reasonable interpretation, since all the speech audio data frames are forwarded to the destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client as recited by the independent claims.

For the second configuration, the Applicant alleges that, "the rate conversion module 130 converts the discontinuous stream of data from the DTX-enabled transmitter into a continuous stream of data for the non-DTX receiver" *(see Applicant's remarks, page 9 lines 22-24).* However, it is noted that the frames that are inserted to provide a continuous stream of data contain background information such as energy level and spectrum represented *(see Benyassine, column 9 lines 14-17).* In contrast, frames containing speech audio data received by the conversion module 230 are transmitted to the receiver 220 *(see Benyassine, column 9 lines 1-5).* Therefore, using the broadest reasonable interpretation, since all the speech audio data frames are forwarded to the

destination without being dropped, all the audio data received by the second client is equal to all the audio data transmitted by the first client as recited by the independent claims.

For at least the reasons provided above, the Applicant's arguments regarding the independent claim are not persuasive. The Applicant further argues that since the dependent claims depend on the argued independent claims, they are patentable by virtue of their dependencies. Since the Applicant's arguments regarding the independent claims are not persuasive, the applicant's arguments regarding the dependent claims are also not persuasive.

Conclusion

11. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. *(See form PTO-892).*

Any inquiry concerning this communication or earlier communications from the examiner should be directed to BEN H. LIU whose telephone number is (571)270-3118. The examiner can normally be reached on 9:00AM to 6:30PM.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Ricky Ngo can be reached on (571)272-3139. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

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/Ricky Ngo/
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